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54 Method and device for demodulating high-frequency modulated signals by means of digital filters and digital demodulators and application of the method in a remote control receiver

The modulated baseband signal occurring as a continuous wanted and interfering signal spectrum is first bandlimited with a bandpass filter and then is sampled with a sampling frequency (f_a) which lies below the minimum sampling frequency prescribed by the sampling theorem. The sampling frequency (f_a) is no lower than twice the value of the upper less the lower stopband frequency of the bandlimited wanted signal spectrum and the wanted signal spectrum lies within one period segment ((1)/2f_a). Through this subsampling, a sampling frequency (f_a) of around 3000 Hz is obtained which makes possible the use of an 8-bit microcomputer as digital filter in a ripple control receiver.

Patent Claims

1. Method for demodulating high-frequency modulated signals by means of digital filters and digital demodulators in which the modulated baseband signal having a continuous wanted and interfering signal spectrum is first bandlimited and then is sampled with a certain sampling frequency and is further processed, characterized in that the modulated baseband signal with the continuous wanted and interfering signal spectrum [$X_A(\Omega)$] is bandlimited with an analog bandpass filter (5), resulting in a wanted signal spectrum [$X_A'(\Omega)$] with a lower and an upper stopband frequency (Ω_{min} and Ω_{max}, respectively) and that a sampling frequency (f_a) is selected, which is not smaller than twice the value of the difference between the upper and the lower stopband frequency ((f_a>2(Ω_{max} - Ω_{min}), with the wanted signal spectrum lying within one period section ($m\frac{1}{2}f_a$) at least up to spectral portions of interfering signals.

2. Method according to Claim 1 characterized in that the signal sampled with the sampling frequency (f_a) is digitally filtered and demodulated in digital manner with a clock frequency corresponding to the sampling frequency.

3. Method according to Claim 2 characterized in that upon the occurrence of spectral portions of interfering signals outside of the aforementioned period segment ($m\frac{1}{2}f_a$), the filter (8), for the digital filtering of the sampled signal, is designed such that said portions come to lie in its stopband.

4. Method according to Claim 3 characterized in that spectral portions of interfering signals not suppressed through the digital filter (8) are attenuated through the analog bandpass filter (5).

5. Device for carrying out the method according to Claim 1 with a filter for bandlimiting the modulated signal and with means for sampling the bandfiltered wanted signal, characterized in that the filter is formed by an analog bandpass filter (5) of the second order, and that the sampling frequency (f_a) is selected such that on the one hand it is not smaller than twice the value of the difference between the upper and lower stopband frequency of the wanted signal limited by the bandpass filter and on the other hand the wanted signal spectrum comes to lie within a period ($m\frac{1}{2}f_a$) of the frequency axis.

6. Device according to Claim 5 characterized through a digital filter (8) for filtering the sampled wanted signal which in the case of whole and half values of the sampling frequency (f_a and $\frac{1}{2}f_a$) has a strong attenuation adapted to the interfering signal spectrum and is driven with a clock frequency corresponding to the sampling frequency.

7. Application of the method according to Claim 1 in a remote control receiver, in particular a ripple control receiver with a digital filter for filtering out amplitude sampled single-tone control signals from the low-voltage network, characterized in that an 8-bit microcomputer is used as digital filter (8) and is sampled with a clock frequency (f_a) which lies below the minimum sampling frequency specified by the sampling theorem.

8. Application according to Claim 7 characterized in that the sampling frequency (f_a) for sampling the received signal with the control frequency f_s, and thus the clock frequency, is selected such that it fulfills the condition $f_a = 3/2\ f_s$ within a band range of ±20-30%.

9. Application according to Claim 8 characterized in that the passbands ($f_a - f_s$) and ($f_a + f_s$) of the analog bandpass filter of the filter system plus digital filter (8) are equally attenuated by the analog bandpass filter (5).

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The invention concerns a method for demodulating high-frequency modulated signals by means of digital filters and digital demodulators in which the modulated baseband signal having a continuous wanted and interfering signal spectrum is first bandlimited and then is sampled and further processed with a specified sampling frequency.

Digital filters and demodulators are currently widely used and among other things serve to filter and demodulate modulated, time-discrete digital signals by means of computers, for example microcomputers or signal processors. The underlying technology is known under the designation "digital signal processing;" see for example the book *Theory and Application of Digital Signal Processing* by L.R. Rabiner and B. Gold, Prentice Hall, New Yersey [sic].

In the theory of digital signal processing, a fundamental requirement concerning the sampling or clock frequency is the so-called sampling theorem. It says that the minimum sampling frequency with which a continuous signal can still be sampled must be at least twice as high as the highest frequency which occurs to a noticeable extent in the spectrum of the signal.

In the book *Halbleiter-Schaltungstechnik* [Semiconductor Circuit Engineering] by U. Tietze and Ch. Schenk, Springer Verlag Berlin, Heidelberg, New York, 1980, it is demonstrated that a time-discrete signal (e.g., a sampled signal with the period $T_a = 1/f_a$)

has a spectrum which is periodic in f_a and that as a result of this periodicity the continuous spectrum $X_A(\Omega)$ must be limited to $|\Omega| \leq \frac{1}{2}f_a$. If this requirement is ignored, a so-called "aliasing" effect occurs as a result of which those portions of the continuous spectrum which are higher than $\frac{1}{2}f_a$ slide into lower frequency ranges and cause interference there.

The height of the sampling frequency together with the length of the processing program determines the minimum computing speed of the computer which must be able to process the entire signal processing program between two samples. Since the computing speed of microcomputers and signal processors is limited, the use of digital signal processing is very often limited thereby.

Through the invention, a method should now be proposed which significantly increases the application possibilities of signal processing in that it makes possible the use of microcomputers with a computing speed which until now was insufficient for digital signal processing.

This object is realized according to the invention in that the modulated baseband signal with the continuous desired and interfering signal spectrum is bandlimited with an analog bandpass filter, resulting in a desired signal spectrum with a lower and an upper stopband frequency and that a sampling frequency is selected which is not lower than twice the difference between the upper and the lower stopband frequency, with the desired signal spectrum lying at least up to spectral portions of interfering signals within one period segment.

The invention is based on the new recognition that said limit of the continuous spectrum $X_A(\Omega)$ cannot only be introduced in the frequency range f_a, but also in that frequency segment of $|m\frac{1}{2}f_a|<|\Omega_A| < |(m + 1)\frac{1}{2}f_a|$ where m = 1, 2, 3, ..., with all spectra which are limited in this manner resulting after sampling in an unambiguous periodic spectrum. In this process, the otherwise so disruptive "aliasing" effect is useful in that the spectra $X_A(\Omega)$ are repeated periodically by $\pm m \cdot f_a$. A spectrum lying in the upper frequency range according to the above inequality is thus mixed down unchanged into the range $|f| \leq \frac{1}{2}f_a$ through "subsampling" with f_a.

The method according to the invention thus makes it possible under certain conditions and with the intervention of the stated measures to sample, filter, and demodulate high-frequency modulated signals with lower sampling frequencies than has been acceptable in the past on the basis of the sampling theorem.

The invention further concerns a device for carrying out said method with a filter for bandlimiting the modulated signal and with means for sampling the bandfiltered wanted signal.

The device according to the invention is characterized in that the filter is formed by an analog bandpass filter of the second order and that the sampling frequency is selected such that on the one hand it is not less than double the difference of the upper and the lower stopband frequency of the wanted signal limited by the bandpass filter and on the other hand the wanted signal spectrum comes to lie within one period of the frequency axis.

The invention also concerns an application of said method in a remote control receiver, in particular in a ripple control receiver, with a digital filter for filtering out amplitude-sampled control signals from the low-voltage network.

The application according to the invention of the method is characterized in that an 8-bit microcomputer is as the digital filter used and that it is sampled with a clock frequency which lies below the minimum sampling frequency required by the sampling theorem.

Ripple control receivers need high quality narrow-band filters in order to filter amplitude-sampled control signals out of the low voltage network. In the past, by way of example, a two-stage digital filter of the fourth order (CH-PS 559 983) which, while it can solve the task, is however significantly more expensive than a microprocessor, for example an 8-bit microcomputer which currently has the greatest practical significance. An 8-bit microcomputer, however, requires computing time of approximately 300 μs for the required calculating operations (approx. 9 multiplications with filter constants, 8 additions with overflow monitoring operations and 8 registration manipulations) and therefore can be clocked, at most, with a clock frequency of 3300 Hz. Since control frequencies up to 2000 Hz occur, however, and accordingly a sampling frequency of at least 4000 Hz is required according to the sampling theorem, in the past there were problems with the computing speed, and 8-bit microcomputers could not be used as digital filters for ripple control receivers.

With the method according to the invention, this task can now for the first time be solved in that it emits lower sampling frequencies in the present case causing the microcomputer to have sufficient computing time.

The invention will be explained in greater detail below with the aid of an exemplary embodiment and the drawings.

Figures 1,2	show diagrams for explaining the method according to the invention,
Figure 3	shows a block diagram of a selective receive section of a ripple control receiver, and
Figures 4,5	show diagrams for explaining functioning.

Figures 1 and 2 show diagrams for explaining the method according to the invention in a general manner. Figure 1 shows in line a modulated baseband signal f_m with a given desired and interfering signal spectrum $X_A(\Omega)$. The desired signal spectrum is designated with N and the interfering signal spectrum is designated with S. The signal of line a is first bandlimited according to line b with an analog bandpass $H_A(\Omega)$, resulting in the desired signal spectrum $X_A'(\Omega)$ in accordance with line c with the stopband limit frequencies Ω_{min} and Ω_{max}.

For sampling the desired signal spectrum $X_A'(\Omega)$, a sampling frequency ω_a is now selected which must satisfy the following conditions:

- $\omega_a \geq 2(\Omega_{max} - \Omega_{min})$
- The desired signal spectrum must lie completely within a period segment $m\frac{1}{2}\omega_a$ ($m = 1,2,3,\ldots$). Outside of this period segment, at most portions of interfering signals which are still spectral may occur. However, this is permissible only if it is ensured that these portions come to lie in the stopband of downstream digital filters.

It can be seen from line c that sampling frequency ω_a may also lie below the occurring signal frequencies.

Upon fulfillment of these two conditions for sampling frequency ω_a, the periodic spectrum $X(e^{j\omega T})$ of the signal sampled with ω_a depicted in line d is obtained. It can be seen from line d that the complete information is contained in each period Π/T.

The signal sampled with ω_a can now be further processed according to known principles of digital signal processing. In particular, it can be digitally filtered—with the clock frequency ω_a—and demodulated in digital manner. Line e shows the frequency response of a digital filter $H(e^{j\omega T})$ for suppression of carrier frequency.

In Figure 2, a depiction is selected for explaining the method according to the invention as in Chapter 2.12 "Relation Between Continuous and Discrete Systems" of the book *Theory and Application of Digital Signal Processing* by L.R. Rabiner and B. Bold, Prentice Hall, New Yersey [sic]. The method is based on the recognition that the limitation of the continuous spectrum $X_A(\Omega)$ described in this chapter can be introduced not only in the frequency range $|\Omega| \leq \frac{1}{2}\,\Pi/T$ or $|\Omega| \leq \frac{1}{2}\,\omega_a$, but also in any frequency segment of $|m\frac{1}{2}\omega_a| < |\Omega_a| < |(m+1)\frac{1}{2}\omega_a|$ where m = 1, 2, 3, …. As is depicted in lines a and b of Figure 2 for m = 2, all spectra limited in this manner result after sampling in an unambiguous periodic spectrum.

In addition it was recognized that in this form of the "aliasing" effect which otherwise was so disruptive becomes useful in that according to formula (2.65) of said chapter, the spectra $X_A(\Omega)$ are periodically repeated by $\pm m2\Pi/F$ or $\pm m\omega_a$. This

means that a spectrum lying in an upper frequency range $|m\frac{1}{2}\omega_a|<|\Omega_a| < |(m + 1)\frac{1}{2}\omega_a|$ is mixed down into the range $|\omega|\leq \frac{1}{2}\omega_a$ unchanged through "subsampling" with ω_a.

In lines c and d of Figure 2, the relationships for m = 1 are depicted. As is shown by a comparison of lines a and b for m = 2 on the one hand and of lines c and d for m = 1 on the other hand, the fact should be noted that depending on whether m is even or odd, the positive or the negative analog frequency spectrum will appear in the range $\omega< \frac{1}{2}\omega_a$.

The sampling theorem must naturally be observed for signal further processing. This requirement, however, is automatically fulfilled if the further signal processing takes place in clock ω_a. In the event that the negative spectrum is further processed, it must be taken into consideration that the frequencies are reflected at $\frac{1}{2}\omega_a$. This as a rule does not present any problems in AM systems and FM systems for digital data transmission (e.g., FSK systems). In audio applications, the spectrum naturally may not be mixed under reversed, to the extent in this case frequency shifts are permissible at all.

The described method is particularly well suited for realizing digital filters in ripple control receivers. As is known, ripple control receivers require high quality narrow-band filters in order to be able to filter out amplitude keyed control signals. In European patent application 83 105 834.2 (publication number 0 105 087), a two-stage digital filter of the fourth order is introduced which fundamentally can solve this task.

If it is desired, however, to implement the digital filter as a favorably priced 8-bit microcomputer, there are problems with the computing speed and with the clock frequency. For on the one hand the necessary computing operations (approx. 9 multiplications with filter constants + 8 additions with overflow monitoring operations + 8 calibration manipulations) require a calculating time of approx. 300 μs so that the digital filter can be clocked at most with a clock frequency of approx. 3300 Hz while on the other hand the sampling theory requires a sampling frequency of at least 4000 Hz as a result of the control frequencies which occur up to 2000 Hz. This means that the digital filter cannot be operated with the sampling frequency prescribed by the sampling theorem.

Figure 3 shows the block diagram of a selective receive section of a ripple control receiver with which with the use of the method according to the invention the task of realization of the required digital filter as an 8-bit microcomputer can be solved.

The selective receive section 1 designated in Figure 3 serves, as is known, to selectively receive from the frequency mix offered from the network a remote control signal with the signal frequency f_s and to emit a pulse sequence corresponding to the remote control commands. The configuration and function of a ripple control receiver are presupposed as known; reference in made in this connection to the aforementioned European patent application 83 105 834.2 and to CH-PS 559 983.

Receive section 1 has an input terminal 2 which is connected at a connection point 3 to a current conductor 4 on which signal frequency f_s is superimposed. The input voltage at the input terminal 2

is carried to a prefilter 5 downstream of which an analog/digital converter 7 and a digital filter 8 are arranged. Arranged behind digital filter 8 is an AM demodulator 9, the output of which is connected to output terminal 10 of receive section 1. Receive section 1 also contains frequency generator 6 having an oscillator quartz for generating the clock frequency for the individual stages of receive section 1. The clock frequency could also be derived from the network by means of a regulation circuit designated as PLL.

Receiver section 1 and its function will now be explained with the aid of Figures 3 through 5, with Figures 4 and 5 showing the signal curves in the individual stages of receive section 1; Figure 4 shows in line a the limitation of the received signal spectrum with prefilter 5 (Figure 3) and in line b the digital spectrum of the sampled signal. Figure 5 shows in line a the filter characteristic of digital filter 8 (Figure 3) in line b the spectrum of the output signal of digital filter 8, in line c the frequency response of amplitude of the filter chain prefilter 5 + digital filter 8, and in line d the attenuating of interfering passbands of the filter chain through prefilter 5.

Prefilter 5 is formed through an analog bandpass filter of the second order with a quality $Q > 15$. In accordance with Figure 4, line a, at f_a and $\frac{1}{2}f_a$, it has an attenuation of –20 dB and limits the received wanted and interfering signal spectrum of control frequency f_s. The clock frequency of frequency generator 6, which corresponds to sampling frequency f_a of A/D converter 7, is selected such that wanted signal spectrum $X_A(jf)$ comes to lie in one period $m\frac{1}{2}f_a$ of the frequency axis. Outside of this period, interfering frequencies are strongly enough attenuated so that digital filter 8 in the passband is not disturbed through original or down-mixed interfering frequencies. According to Figure 4,

line a, sampling frequency f_a is 3000 Hz, one-half sampling frequency $\frac{1}{2}f_a$ thus is 1500 Hz, and the signal lies in the frequency period between $m\frac{1}{2}f_a$ and $(m+1)\frac{1}{2}f_a$, where m=1.

After the sampling of the band filtered network signal, a digital spectrum $X(e^{j2\Pi fT})$ according to Figure 4, line b is obtained. Dashed curve A shows the sum of all overlapping periodic spectra. Since the bandpass filter according to line a has only a finite attenuation (-20 dB), some interfering "aliasing" still occurs.

The output signal of A/D converter 7 (Figure 3) is filtered with digital filter 8 which by way of example can be of the type described in European patent application 83 105 834.2 and has a filter characteristic according to Figure 5, line a. Since this filter at $\frac{1}{2}$ f_a and f_a attenuates strongly (-20 dB), the interfering "aliasing" effects are strongly suppressed.

Line b of Figure 5 shows the spectrum of output signal of digital filter 8 (Figure 3): $Y(e^{j2\Pi fT}) = X(e^{j2\Pi fT}) \cdot H(e^{j2\Pi fT})$. The spectrum is characteristic in the ranges $f_a - f_s$ and $f_a + f_s$, where new frequencies arise as a result of the periodicity of signal spectrum $X(e^{j2\Pi fT})$ and of filter transmission function $H(e^{j2\Pi fT})$ ("aliasing" through sampling). In the range $f_a - f_s$, the spectrum compared with the original spectrum is reflected to f_a. This, however, does not have any influence on further processing, since only the amplitude of the signal is supposed to be evaluated. In FSK systems, on the other hand, the reflection would have taken into consideration.

The output signal of digital filter 8 is evaluated with respect to amplitude by AM demodulator 9 (Figure 3), which preferably is realized digitally. This evaluation can take place as follows: Digital signal $Y(e^{j2\Pi fT})$ is rectified for each sample; thus the absolute value of Y(nT) is formed, Y(nT) = |Y(nT) [sic]. This absolute value is presented to a digital lowpass filter which is also clocked with f_a and has limit frequency adapted to the frequency of the baseband signal which, however, of course is below $\frac{1}{2}f_a$.

In Figure 5, line c, the realized single-tone transmission function, i.e. the amplitude response of the filter chain of analog bandpass 5 and digital filter 8 (Figure 3) is depicted. The transmission is not frequency true, because if a control frequency f_s is presented to the filter, the fundamental frequency $f_a - f_s$ (dashed line B) appears at its output as a result of subsampling. The same frequency $f_a - f_s$ also appears in the case of feeding in the frequency $f_l = f_a - f_s$, but this frequency f_l is attenuated by 25 dB, which is suggested through point C. This attenuation is achieved solely through prefilter 5 (Figure 3). In like manner the frequency $f_a - f_s$ appears at the output of digital filter 8 (Figure 3) if it is actuated with any frequency $m(f_a \pm f_s)$ where m=1,2,.... The attenuation of all of these periodic frequencies is likewise provided exclusively through prefilter 5.

Since according to the invention the specification for the sampling frequency merely says that the wanted signal spectrum must lie within a frequency period of $\frac{1}{2}f_a$, there is still a degree of freedom in selecting the sampling frequency. In addition it became apparent that interfering spectra outside of frequency period $\frac{1}{2}f_a$ must be attenuated by prefilter 5 only to the extent they are not suppressed through following digital filter 8 (Figure 3).

This leads to the task of optimally configuring sampling frequency f_a together with analog bandpass filter 5 and digital filter 8. In the present case, a particularly suitable solution is for sampling frequency f_a to fulfill the following condition:

$$f_a = 3/2 \cdot f_s$$

In this case it of course follows that the two critical "passbands" of the filter system which are to be attenuated by the analog prefilter, $f_a - f_s$ and $f_a + f_s$, are in a relationship to f_s of 1:2 or 2:1. Thus it is ensured, as is shown in Figure 5, line d, that the analog bandpass filter of second order attenuates the two interfering passbands equally. For the rations 1:2 and 2:1 result in equivalent frequency spacing on the logarithmic frequency scale.

Since in the case of ripple control receivers, control frequency f_s is up to 2000 Hz, a sampling frequency f_a of 3000 Hz is obtained. This sampling and clock frequency is still sufficiently low even for simple 8-bit microcomputers, while the sampling frequency required by the sampling theorem of at least 4000 Hz clearly would be too high. Instances are also conceivable in which the described method will result in a still higher "subsampling gain," for example in the case of high-frequency modulated narrow spectra. They can be limited using a narrow bandpass filter and transformed through subsampling into a lower frequency range and there be fine-filtered.

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[Figure 2]

[Figure 3]
[Figure 4]

[Figure 5]

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[Figure 1]

Period 1 Period 2 Period 3